Clmpto
Klv
04/20/01

1. A speech encoder comprising:

LPC synthesizing means for obtaining a synthesized speech by filtering adaptive excitation vector and stochastic excitation vector stored in an adaptive codebook and stochastic codebook using an LPC coefficients obtained from an input speech;

gain calculating means for calculating gains of said adaptive excitation vector and said stochastic excitation vector and searching code of the adaptive excitation vector and searching code of the stochastic excitation vector using coding distortion between said input speech and said synthesized speech obtained using said gains; and

parameter coding means for performing predictive coding of gains using the adaptive excitation vector and stochastic excitation vector corresponding to the codes obtained, wherein said parameter coding means comprises prediction coefficient adjusting means for adjusting one or more prediction coefficients used for said predictive coding according to one or more states of a previous subframe.

2. (Amended) The speech encoder according to claim 1, wherein when one or more states of a previous subframe are an extremely large value or an extremely small value, said prediction coefficient adjusting means adjusts said prediction coefficients so as to reduce the influence thereof.

3. The speech encoder according to claim 1, wherein said parameter coding means has a codebook including gain vectors of the adaptive excitation vectors, logarithmic gain vectors of the stochastic excitation vectors and coefficients for adjusting the prediction coefficient.

4. The speech encoder according to claim 3, wherein in predicting coding when a product sum between states and prediction coefficients are calculated, prediction coefficient adjustment coefficients corresponding to the states are multiplied.

5. The speech encoder according to claim 1, further comprising storing means for storing said adaptive excitation vector, said stochastic excitation vector and prediction coefficient adjustment coefficient in accordance with each state.

6. (Amended) The speech encoder according to claim 5, wherein when said adaptive excitation vector and said stochastic excitation vector stored in said storing means are updated, said prediction coefficient adjustment coefficients are also updated.

7. A CELP-based speech encoder that performs encoding by decomposing one frame into a plurality of subframes, comprising:

LPC synthesizing means for obtaining a synthesized speech by filtering adaptive excitation vector and stochastic excitation vector stored in an adaptive codebook and stochastic codebook using LPC coefficienta obtained from an input speech;

gain calculating means for calculating gains of said adaptive excitation vector and said stochastic excitation vector; and

parameter coding means for performing vector quantization of the adaptive excitation vector and stochastic excitation vector obtained using coding distortion between said input speech and said synthesized speech and said gains, and further comprising:

pitch analyzing means for performing pitch analyses of a plurality of subframes in the frame respectively, before performing an adaptive codebook search for the first subframe, finding a correlation value and calculating a value most approximate to the pitch period using said correlation values.

8. The speech encoder according to claim 7, further comprising search range setting means for determining a lag search range of a plurality of subframes based on the correlation value and the value most approximate to the pitch period obtained by said pitch analyzing means.

9. The speech encoder according to claim 8, wherein said search range setting means determines a provisional pitch that becomes the center of the search range using the correlation values and the values most approximate to the pitch period obtained by said pitch analyzing means.

10. The speech encoder according to claim 9, wherein the search range setting means sets a lag search section

in a specified range around the provisional pitch.

11. The speech encoder according to claim 8, wherein the search range setting means sets a lag search section by reducing the number of candidates for shortpitch periods.

12. The speech encoder according to claim 8, wherein the search range setting means performs a lag search within a set range during an adaptive codebook search.

13. A computer-readable recording media storing a speech encoding program, an adaptive codebook storing past used for synthesising excitation vector signals and a stochastic codebook storing a plurality of stochastic excitation vectors, said speech encoding program comprising the steps of:

obtaining a synthesized speech by filtering adaptive excitation vector and by filtering stochastic excitation vector stored in said adaptive codebook and in said stochastic codebook using decoded LPC coefficients obtained from an input speech;

calculating gains of said adaptive excitation vector and said stochastic excitation vector;

performing vector quantization on the adaptive excitation vector and stochastic excitation vector determined using coding distortion between said input speech and said synthesized speech, and said gains, wherein said vector quantization step further comprising the steps of:

determining a quantization target vector based on

coding distortion between a plurality of quantization target vector and prediction coefficient used for predictive coding; and

adjusting said prediction coefficients according to one or more states of on or more previous subframe.

14. A computer-readable recording medium storing a speech encoding program, an adaptive codebook storing past used for synthesising excitation vector signal and a stochastic codebook storing a plurality of stochastic excitation vector, said speech coding program comprising the steps of:

obtaining a synthesized speech by filtering adaptive excitation vector and stochastic excitation vector stored in said adaptive codebook and said stochastic codebook using decoded LPC coefficients obtained from an input speech;

calculating gains of said adaptive excitation vector and said stochastic excitation vector;

performing vector quantization on the adaptive excitation vector and stochastic excitation vector determined using coding distortion between said input speech and said synthesized speech; and

calculating correlation values by performing pitch analysses of a plurality of subframes in the processing frame before performing an adaptive codebook search of the first subframe and calculating a value most approximate to the pitch period using said correlation values.

15. (Added) A speech encoder comprising a dispersed-pulse codebook that generates a vector by convoluting a vector containing one or more one non-zero elements (elements other than non-zero elements have values of zero) and a fixed waveform called a "dispersion pattern", wherein said dispersed-pulse codebook has a configuration different from that of the dispersed-pulse codebook on the speech decoder side.

16. (Added) The speech encoder according to claim 15, wherein the dispersion pattern storage section, which is a component of the dispersed-pulse codebook, stores dispersion patterns different from those stored in the dispersion pattern storage section on the speech decoder side.

17. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by simplifying and selecting dispersion patterns stored in the dispersion pattern storage section on the speech decoder side.

18. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by replacing components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side with zero at certain intervals.

19. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by replacing

components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side with zero for every N samples (N: natural number).

20. (Added) The speech encoder according to claim 19, wherein the dispersion pattern storage section stores dispersion patterns obtained by replacing components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side with zero for every 1 sample.

21. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by truncating components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side at an appropriate length.

22. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by truncating components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side at a length of N samples (N: natural number).

23. (Added) The speech encoder according to claim 16, wherein the dispersion pattern storage section stores dispersion patterns obtained by truncating components of dispersion patterns stored in the dispersion pattern storage section on the speech decoder side at a half length.

24. (Added) A speech decoder that decodes a speech

signal having a speech code generated by the speech encoder according to claim 15.

25. (Added) A signal processing processor containing a software program that implements the speech encoder according to claim 15.

26. (Added) A signal processing processor containing a software program that implements the speech decoder according to claim 24.

27. (Added) A speech encoding/decoding system comprising a speech encoder and/or a speech decoder each having a dispersed-pulse codebook in configuration different from each other.

28. (Added) The speech encoding/decoding system according to claim 27, wherein the difference in the configuration of the dispersed-pulse codebook between the speech encoder and the speech decoder lies in the shape of dispersion patterns stored in the respective dispersed-pulse codebooks.

29. (Added) The speech encoding/decoding system according to claim 28, wherein the shapes of dispersion patterns of the speech encoder are obtained by simplifying the shape of dispersion patterns of the speech decoder.

30. (Added) The speech encoding/decoding system according to claim 27, wherein the shapes of dispersion patterns of the speech encoder are obtained by replacing components of the dispersion patterns of the speech decoder with zero at appropriate intervals.

31. (Added) The speech encoding/decoding system according to claim 27, wherein the shapes of dispersion patterns of the speech encoder are obtained by replacing components of the dispersion patterns of the speech decoder with zero every N samples (N: natural number).

32. (Added) The speech encoding/decoding system according to claim 31, wherein the shapes of dispersion patterns of the speech encoder are obtained by replacing components of the dispersion patterns of the speech decoder with zero every 1 sample.

33. (Added) The speech encoding/decoding system according to claim 27, wherein the shapes of dispersion patterns of the speech encoder are obtained by truncating components of the dispersion patterns of the speech decoder at an appropriate length.

34. (Added) The speech encoding/decoding system according to claim 27, wherein the shapes of dispersion patterns of the speech encoder are obtained by truncating components of the dispersion patterns of the speech decoder at a length of N samples (N: natural number).

35. (Added) The speech encoding/decoding system according to claim 27, wherein the shapes of dispersion patterns of the speech encoder are obtained by truncating components of the dispersion patterns of the speech decoder at a half length.

36. (Added) A communication base station equipped with the signal processing processor according to claim 25.

37. (Added) A communication terminal equipped with the signal processing processor according to claim 25.

38. (Amended) A radio communication system that connects the communication base station according to claim 36 with [the] a communication terminal [according to claim 37] via a radio network.